In the claims:

Please amend the claims as follows:

1(currently amended). A telephony communications network comprising:

a telephony unit generating a <u>private</u> telephony signaling code;

a translator coupled to the telephony unit, the translator encapsulating the <u>private</u> telephony signaling code in a application layer control protocol message; and

a communications interface coupled to the translator for transmitting the message over a communications network.

2(currently amended). The network of claim 1 further comprising:

one or more second telephony units; and

one or more second translators coupled to the one or more second telephony units, characterized in that the one or more second translators receive the message transmitted over the communications network and decapsulate the <u>private</u> telephony signaling code in the message, further characterized in that the one or more second translators forward the <u>private</u> telephony signaling code to the one or more second telephony units for performing a function in response to the <u>private</u> telephony signaling code.

3(original). The network of claim 2 further comprising a server for routing the message to the one or more second translators.

4(original). The network of claim 3, wherein the server provides a third party service for the telephony unit.

5(original). The network of claim 2, wherein the translator determines the one or more second translators viable for receiving the message and transmitting the message to the viable translators.

6(original). The network of claim 2, wherein the one or more second translators subscribe with the translator for receiving the message.

7(original). The network of claim 2, wherein the one or more second translators receive the message in an out-of-call data transfer.

8(original). The network of claim 1, wherein the telephony unit is a digital telephone.

9(original). The network of claim 1, wherein the telephony unit is a private branch exchange unit.

10(original). The network of claim 1, wherein the session layer control protocol is a Session Initiation Protocol (SIP).

11(original). The network of claim 10, wherein the telephony unit is a SIP user agent.

12 (original). A telephony communications network supporting a session initiation protocol (SIP) session, the network comprising:

a SIP client transmitting and receiving SIP messages during the SIP session; and

a translator coupled to the SIP client, the translator configured to encapsulate and decapsulate private telephony signaling codes in and from the SIP messages for allowing the SIP client access to PBX functionality associated with the private telephony signaling codes.

13(original). The network of claim 12, wherein the translator is an application programming interface.

14(original). A telephony communications network supporting a session initiation protocol (SIP) session, the system comprising:

a telephone appliance transmitting outgoing private telephony signaling codes for accessing associated PBX functionality;

a translator coupled to the telephone appliance, the translator configured to encapsulate the outgoing private signaling codes in outgoing SIP messages transmitted to the PBX via a SIP server providing a SIP service for the telephone appliance.

15(original). The network of claim 14, wherein the SIP service is a call redirection service.

16(currently amended). In a telephony communications network comprising a telephony unit and a translator coupled to the telephony unit, the translator comprising:

a signaling interface receiving a <u>private</u> telephony signaling code;

a processor coupled to the signaling interface, the processor configured to encapsulate the <u>private</u> telephony signaling code in a session layer control protocol message; and

a network interface coupled to the processor for transmitting the message over a communications network.

17(currently amended). The translator of claim 16, wherein the processor is further configured to decapsulate a second <u>private</u> telephony signaling code in a second session layer control protocol message received by the network interface for forwarding to the telephony unit over the signaling interface.

18(original). The translator of claim 16, wherein the session layer control protocol is a Session Initiation Protocol.

19(original). The translator of claim 18, wherein the telephony unit is a SIP user agent.

20(original). The translator of claim 16, wherein the telephony unit is a digital telephone.

21(original). The translator of claim 16, wherein the telephony unit is a private branch exchange unit.

22(currently amended). A method for communication over a telephony network comprising:

generating a private telephony signaling code;

encapsulating the private telephony signaling code in a session layer control protocol message; and

transmitting the message over a communications network.

23(currently amended). The method of claim 22 further comprising:

receiving the message transmitted over the communications network;

decapsulating the private telephony signaling code in the message; and

performing a function in response to the private telephony signaling code.

24(original). The method of claim 22, wherein the transmitting of the message comprises determining one or more viable destinations to which to transmit the message and transmitting the message to the viable destinations.

25(original). The method of claim 22, wherein the transmitting of the message comprises transmitting the message in an out-of-call data transfer.

26(original). The method of claim 22 further comprising transmitting a subscription message for receiving the message.

27(original). The method of claim 22, wherein the session layer control protocol is a Session Initiation Protocol (SIP).

28(original). The method of claim 27 further comprising routing the message over a SIP server